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EXAMINER	
ARMSTRONG, ANGELA A	

ART UNIT	PAPER NUMBER
2654	

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/302,397	Applicant(s) OZAWA, KAZUNORI	
	Examiner Angela A. Armstrong	Art Unit 2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *21 September 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-4 and 6-11* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☒ Claim(s) *6 and 7* is/are allowed.

6)☒ Claim(s) *1-4, 8-11* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. In view of the Appeal Brief filed on September 21, 2004, PROSECUTION IS HEREBY REOPENED.

To avoid abandonment of the application, appellant must exercise one of the following two options:

(1) file a reply under 37 CFR 1.111 (if this Office action is non-final) or a reply under 37 CFR 1.113 (if this Office action is final); or,

(2) request reinstatement of the appeal.

If reinstatement of the appeal is requested, such request must be accompanied by a supplemental appeal brief, but no new amendments, affidavits (37 CFR 1.130, 1.131 or 1.132) or other evidence are permitted. See 37 CFR 1.193(b)(2).

2. Applicant's arguments, see Appeal Brief, page 18, last paragraph, filed September 21, 2004, with respect to claims 6 and 7 have been fully considered and are persuasive. The rejection of claims 6 and 7 has been withdrawn.

3. Applicant's arguments filed regarding claims 1-4, and 8-11 have been fully considered but they are not persuasive.

Applicant argues it is unnecessary to transmit mode discrimination information to the reception side. In response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the features upon which applicant relies (i.e.,

mode discrimination information is not transmitted to the reception side) are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993). The claims uses claim language of "comprising" and "including at least", this claim language indicates other features or limitations are included, and thus the claims do not specifically teach not allowing for mode discrimination information being transmitted, and the prior art combination which provides transmission of mode information reads on the claim limitations.

Applicant argues the modifications of Kleijn as proposed by the Examiner is not fairly taught by the references and it is not at all clear that such modifications would result in a working coder/decoder and that the Examiner has identified out of context features in Ozawa and say that they could be used to modify Kleijn.

In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988) and *In re Jones*, 958 F.2d 347, 21 USPQ2d 1941 (Fed. Cir. 1992). In this case, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder and provides for mode selection such that coding methods and codebooks are changed to improve coding efficiency as

well as to reduce codebook size. Further, the test for obviousness is not whether the features of a secondary reference may be bodily incorporated into the structure of the primary reference; nor is it that the claimed invention must be expressly suggested in any one or all of the references. Rather, the test is what the combined teachings of the references would have suggested to those of ordinary skill in the art. See *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 1-4 and 8-11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kleijn et al (US Patent No. 5,704,003) in view of Ozawa et al, "M-LCELP Speech Coding at 4KBPS", Acoustics, Speech, and Signal Processing, 1994. ICASSP-94, 1994 IEEE International Conference on, Volume: 1, 19-22 April 1994, Page(s): I/269 -I/272 vol.1.

5. Regarding claim 1, Kleijn et al teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3.

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn also discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of pulses and amplitudes and searches code vectors stored in the codebook and delays or shift amounts so as to output a combination of code vector and shift amount that minimizes distortion at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

6. Regarding claim 2, Kleijn discloses spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn disclose an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62 and a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in

accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discriminating section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

7. Regarding claims 3, 8 and 11, Kleijn discloses a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn discloses an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from the discriminating section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

8. Regarding claim 4, Kleijn teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teaches, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

9. Regarding claims 9-10, Kleijn and Ozawa et al teach everything as claimed in claim 8. Additionally, at col.7, lines 2-26, Kleijn discloses determination of the time shift amount is based on a value that minimizes a certain criteria, which reads on "sound source quantization uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode."

Allowable Subject Matter

10. Claims 6 and 7 are allowed.

Conclusion

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 703-308-6258. The examiner can normally be reached on Monday-Thursday 7:30-5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (703) 305-9645. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Angela A. Armstrong
Examiner
Art Unit 2654

AAA
February 19, 2005

Angela Armstrong